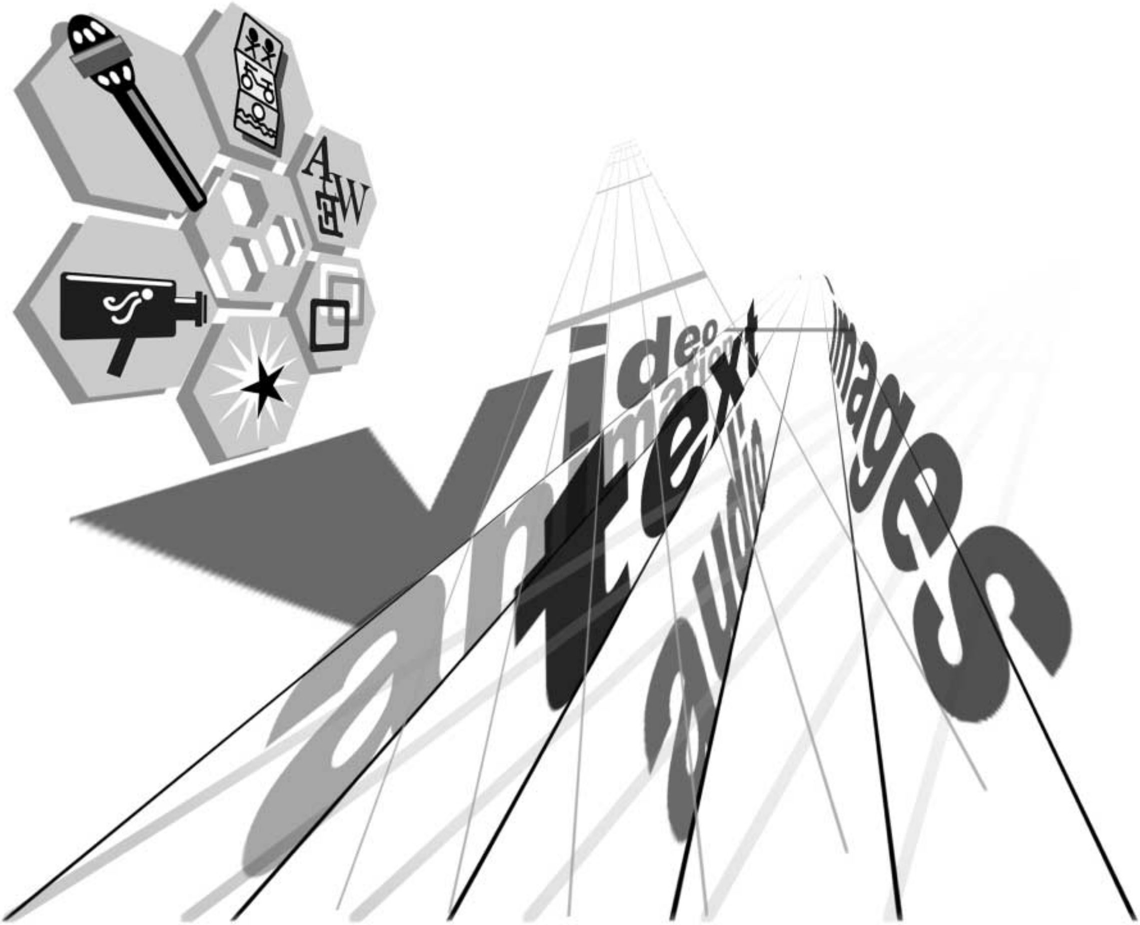
WORKING WITH REALPRODUCER 8 CODECS

RealNetworks Technical Blueprint Series

19 May 2000



RealNetworks, Inc.
2601 Elliott Avenue
Seattle, WA 98121 USA

http://www.realnetworks.com

CONTENTS

TABLES

FIGURES

WORKING WITH REALPRODUCER 8 CODECS

Intended for Web professionals, intranet managers, and streaming media enthusiasts, this document explains RealAudio and RealVideo codecs in depth. It will help you use RealProducer or RealProducer Plus to create RealAudio and RealVideo clips that stream well and encode your source material faithfully.

What this Document Teaches You

You'll learn the following from this document:

- What SureStream is, and why it won't work with a Web server.
- What a duress stream is.
- What a codec is.
- The bit rates for RealAudio default codecs.
- How and when to change the RealAudio codecs for a target audience.
- How and when to change the RealAudio codecs used with RealVideo.
- When to use a RealVideo G2 codec instead of the RealVideo 8.0 codec.
- What RealVideo's Scalable Video Technology (SVT) does.
- How to modify RealVideo's maximum frame rate and streaming bit rate.
- When to use the RealVideo prefilters.
- How variable bit rate clips differ from constant bit rate clips.
- How to read RealProducer's encoding statistics.

Requirements for Using this Document

Before you use this document, you'll need the following:

- Version 8 of RealProducer Basic or RealProducer Plus

  This document is specific to version 8 of RealProducer. If you are using an earlier version of RealProducer, RealNetworks recommends that you upgrade to version 8. You can purchase RealProducer Plus or download the RealProducer Basic from **http://www.realnetworks.com/products/index.html**.

- Experience running RealProducer

  This document does not provide step-by-step instructions for using RealProducer. It assumes that you are familiar with the RealProducer user interface and the procedures for encoding clips, which are described in the RealProducer online help, as well as *RealProducer Plus User's Guide*.

- Basic Internet multimedia knowledge

  You should be familiar with digitized audio and video, as well as the basic production techniques for creating streaming clips. You can find tips for these procedures in the audio and video production chapters of *RealSystem G2 Production Guide*, which is included with RealProducer Plus and available at **http://service.real.com/help/library/encoders.html**.

Note

Although many tips described here apply to Internet broadcasting, this document focuses on creating clips for on-demand delivery.

Changes in this Document Release

This technical blueprint is an update to *Working with RealProducer 7 Codecs*. If you are familiar with the earlier document, this document has only a few changes:

- This document discusses the new RealVideo 8.0 codec in "RealVideo 8.0" on page 30.
- The tables in this document reflect values for the 64 Kbps single ISDN target audience, which was reinstated in RealProducer 8.

Choosing SureStream or Single Rate Encoding

When you encode a clip with RealProducer, the most important decision you make is whether to create a single-rate clip or use SureStream to generate a multi-rate clip. This choice is simple:

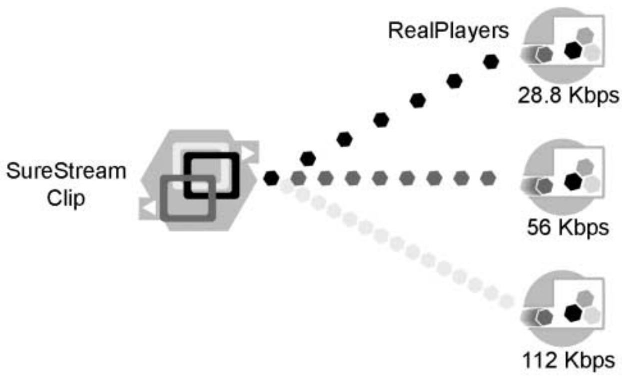
- To stream a clip with RealServer, choose SureStream.
- To host a clip on a Web server, choose the single rate option.

However, to use RealProducer effectively, you should understand what SureStream is. This will help you later if you decide to change the RealProducer encoding defaults.

What is SureStream?

Introduced with RealSystem G2, SureStream technology gives the best possible playback experience for different types of network connections. With SureStream, you can encode your source file for optimum streaming over, for example, 28.8 Kbps modems, 56 Kbps modems, and 112 Kbps dual ISDN connections, as illustrated in Figure 1. Users with 28.8 Kbps modems receive the stream at the lowest bit rate, while users with faster connections get a stream with better quality because of the extra bandwidth. RealProducer Basic encodes two speeds per clip, RealProducer Plus up to eight speeds.

Figure 1: SureStream Clip Encoded for Several Bandwidths



A SureStream clip can also "downshift" when a connection gets bogged down. Suppose a viewer starts out receiving a higher bit rate stream through a 56 Kbps modem. If the connection starts to sputter as the network gets crowded, RealServer automatically downshifts to a lower bit rate stream. The audio or video quality drops with the lower rate stream, but the clip keeps playing.

When the interference clears, RealServer upshifts to the higher rate stream. RealPlayer doesn't need to rebuffer during this shifting.

Why SureStream won't Work with Web Servers

SureStream works only with clips streamed with RealServer. Because SureStream wraps several streams into a single clip, a server has to know how to extract just one stream from the clip. RealServer can do this, but Web servers, which don't know the details of SureStream, send down all the encoded data for a SureStream clip rather than just one stream.

Duress Streams

Although you can encode a clip for just one connection speed with SureStream, you should never do this when delivering a clip with a Web server. Even when encoded for just one speed, a SureStream clip contains "duress" streams that let it downshift during network turbulence. For example, a SureStream music clip encoded only for 28.8 Kbps has duress streams at 16 and 11 Kbps. A Web server downloads all these streams, wasting bandwidth.

Multiple Bandwidth Choices Through SMIL

When using a Web server, you can provide different bandwidth choices by encoding your source file several times for different connection speeds. You then provide multiple links on your Web page (one for each clip), or use SMIL's `<switch>` tag so each RealPlayer chooses the right clip automatically. SMIL isn't as flexible as SureStream, and you won't get downshifting, but using SMIL makes it easier for your viewers to get the best clip for them.

Additional Information

See the SMIL chapter in *RealSystem G2 Production Guide* for instructions on using a `<switch>` tag.

SureStream File Size

Because each SureStream stream increases the clip's file size, an encoded clip may be larger than its source file, even though the clip is compressed. If the clip's file size is an issue, limit the SureStream streams to just the essential speeds. Table 1 gives ballpark figures for file space required for each minute of a RealAudio or RealVideo stream.

When using SureStream to encode RealAudio or RealVideo, add the figures from Table 1 for each connection speed you encode, then multiply by the

Table 1: Approximate Size for RealAudio and RealVideo Clips

Target Audience	RealAudio	RealVideo
28.8 Kbps modem	0.15 Megabytes/minute	0.15 Megabytes/minute
56 Kbps modem	0.24 Megabytes/minute	0.25 Megabytes/minute
64 Kbps single ISDN	0.32 Megabytes/minute	0.33 Megabytes/minute
112 Kbps dual ISDN	0.47 Megabytes/minute	0.59 Megabytes/minute
Corporate LAN	0.70 Megabytes/minute	1.1 Megabytes/minute
256 Kbps DSL/cable modem	0.70 Megabytes/minute	1.65 Megabytes/minute
384 Kbps DSL/cable modem	0.70 Megabytes/minute	2.56 Megabytes/minute
512 Kbps DSL/cable modem	0.70 Megabytes/minute	3.30 Megabytes/minute

length in minutes the source file plays. Table 1 shows that if you encode a RealAudio clip for 28.8 Kbps and 56 Kbps modems, for example, the clip takes about 0.39 Megabytes of disk space per minute (0.15 + 0.24). If the audio is five minutes long, the clip will be about 1.95 Megabytes in size (0.39 x 5).

The figure you get is an approximation based on using the RealProducer default settings. The actual file size varies depending on the combination of streams you've chosen, the type of audio, and whether you've changed the defaults. If you encode for just one connection speed, the file will likely be larger than you calculated because of the duress streams. When you encode RealVideo for multiple connections, the file size may be lower than you calculated because different RealVideo streams may use the same RealAudio stream for a soundtrack.

Working with RealAudio Codecs

All streaming audio and video clips use some type of codec, which is short for "coder/decoder." Codec software tells a computer how to compress or decompress a clip. RealProducer uses RealAudio and RealVideo codecs to convert audio and video input to RealAudio and RealVideo clips that stream at a certain bandwidth. On the receiving end, RealPlayer uses the same codecs to expand the streaming clips into digitized audio and video data that the computer can play.

RealAudio employs a series of codecs, each of which uses a precise amount of bandwidth when a clip streams. One RealAudio codec compresses mono music for a 28.8 Kbps modem. Another compresses stereo music for that same modem speed. This set of codecs is different from the set used to compress

music for, say, cable modems. "RealAudio Codec Reference" on page 16 lists the many RealAudio codecs RealProducer uses.

When you encode RealAudio with RealProducer, you select target audience speeds and indicate the audio type, whether voice, voice with music, mono music, or stereo music. By doing this, you tell RealProducer which RealAudio codec or codecs to use. That's why you need to answer RealProducer's questions correctly. Encoding with the wrong codec may degrade a clip's sound quality. Using a codec intended for voice on a music clip forces, as it were, a square peg through a round hole.

RealAudio squeezes files down by, *in part*, throwing out data, which makes RealAudio a *lossy* compression format. RealAudio doesn't delete data indiscriminately, though. It first jettisons portions you can't hear, such as very high and very low frequencies. Next, it removes as much data as needed while keeping certain frequencies intact. Your answer to the audio type question (voice, music, or a combination) determines which frequencies stay and which go. Voice encoding favors frequencies in the normal human speaking range. Music encoding retains a broader frequency range.

Although RealProducer is savvy about what audio data it throws out, you can't escape the reality that the lower the connection speed, the more data gets ejected, and the cruder the sound quality becomes. At low bandwidths, you get roughly the quality of an AM radio broadcast. With faster connections, you can encode music with FM-quality sound. And at the high speeds of DSL, cable modems, and LANs, RealAudio rivals CD playback.

Encoding with the RealAudio Standard Defaults

Out of the box, RealProducer is set up to use a specific set of RealAudio codecs based on your choice of:

- target audience speed (dial-up modem, DSL/cable, and so on), and
- audio type (voice, voice with music, mono music, or stereo music).

Table 2 shows the bit rates at which RealProducer encodes a RealAudio file based on your choice of connection speed and audio type. With voice-only clips encoded for a 28.8 Kbps modem, for example, you get a 16 Kbps streaming clip. With mono music, though, you get a 20 Kbps streaming clip.

Table 2: RealAudio Standard Default Bit Rates

<table>
<tr><th>Target Audience</th><th>Voice Only</th><th>Voice and Music</th><th>Mono Music</th><th>Stereo Music</th></tr>
<tr><td>28.8 Kbps modem</td><td rowspan="2">16 Kbps</td><td>16 Kbps</td><td>20 Kbps</td><td>20 Kbps</td></tr>
<tr><td>56 Kbps modem</td><td rowspan="2">32 Kbps</td><td>32 Kbps</td><td>32 Kbps</td></tr>
<tr><td>64 Kbps single ISDN</td><td>32 Kbps</td><td>44 Kbps</td><td>44 Kbps</td></tr>
<tr><td>112 Kbps dual ISDN</td><td rowspan="5">64 Kbps</td><td rowspan="5">64 Kbps</td><td rowspan="5">64 Kbps</td><td>64 Kbps</td></tr>
<tr><td>Corporate LAN</td><td rowspan="4">96 Kbps</td></tr>
<tr><td>256 Kbps DSL/cable modem</td></tr>
<tr><td>384 Kbps DSL/cable modem</td></tr>
<tr><td>512 Kbps DSL/cable modem</td></tr>
</table>

Notice that RealAudio doesn't use all of a connection's available bandwidth. For 28.8 Kbps modems, RealAudio uses 16 or 20 Kbps, reserving some bandwidth for network overhead. You can never push a full 28.8 Kbps of usable data through a 28.8 Kbps modem. Although the same is true for all connections, higher-speed connections need proportionally less bandwidth for overhead than analog modems, which are more prone to transmission errors.

At low speeds, RealAudio uses most of the connection bandwidth to make the audio quality as high as possible. At faster speeds, RealAudio uses less of the connection's total bandwidth. The fastest RealAudio codec is 96 Kbps, which is less than half the usable bandwidth of a 256 Kbps connection. Note, too, that every RealAudio codec has a specific speed. There are 64 Kbps and 96 Kbps codecs, for example, but nothing in-between. These are the factory settings, and you can't make RealAudio stream at, say, 75 Kbps.

Choosing SureStream Rates with RealProducer Basic

Getting a handle on the RealAudio codecs used with RealProducer's target audience speeds helps you use SureStream effectively. If you use RealProducer Basic, you get just two audience choices, so you should make the most of them. Table 2 tells you *not* to choose the 28.8 and 56 Kbps modem audiences when encoding a voice-only clip, for instance. If you do, RealProducer creates a single 16 Kbps stream for both audiences. It's better to choose one of the modem settings (it doesn't matter which) and one of the higher bandwidth settings. This gives you a clip with a 16 Kbps stream for slower connections, and a 32 or 64 Kbps stream for faster connections.

With music, though, you need to determine which audiences to leave out. For mono music, a 28.8 Kbps modem gets 20 Kbps of sound, while a 56 Kbps modem gets 32 Kbps. That extra 12 Kbps of bandwidth noticeably improves sound quality. So if modem audiences are important, encode with the two modem settings. If you want to provide higher-quality clips for faster connections, choose one modem audience to get a slower clip (16, 20, or 32 Kbps), and a higher-speed connection to get a faster clip (44, 64, or 96 Kbps). Just remember not to choose two audiences that get the same stream.

Choosing SureStream Rates with RealProducer Plus

If you purchase RealProducer Plus, you can encode RealAudio for all eight target audience speeds. It doesn't hurt to choose all the audiences, but you won't ever get eight distinct and separate streams. As shown in Table 2 above, many target audiences share the same stream. RealProducer is smart enough to know that if you choose different audiences that use the same codec, you need just one stream that uses that codec. It won't waste file space and processor time by stuffing two identical streams into the same clip.

In Table 2, notice that RealProducer Plus gives you, at most, five SureStream streams per RealAudio clip. Aren't you're supposed to get up to eight? This isn't a marketing ploy, as you'll see with RealVideo, which differs greatly in encoding speed for each target audience. And as described in "Changing RealAudio Defaults" on page 12, you can modify the RealAudio default settings to get different streaming rates for each target audience.

Reducing Streaming Clip Size

Just because RealProducer Plus lets you encode for all eight target audiences, that doesn't mean you have to. Other factors such as clip size might matter. Each SureStream stream increases the clip's size as shown in Table 1 on page 5. If the audio file is large, the encoded clip size can escalate dramatically. The bigger the clip, the more hard disk space it eats up, and the longer it takes to transfer to RealServer. If you need to reduce the clip size, Table 2 on page 7 helps you intelligently select which connection speeds to leave out.

Suppose you're encoding stereo music. You want to cut down the file size, and the music quality is fine at medium bandwidths. Table 2 tells you to leave out the LAN and DSL/cable modem options. Just leaving out one of these options won't help because they all use a 96 Kbps stream. Leaving them all out eliminates the 96 Kbps stream, reducing the clip size quite a bit. Users with these fast connections get the 64 Kbps stream, which still has good quality. If

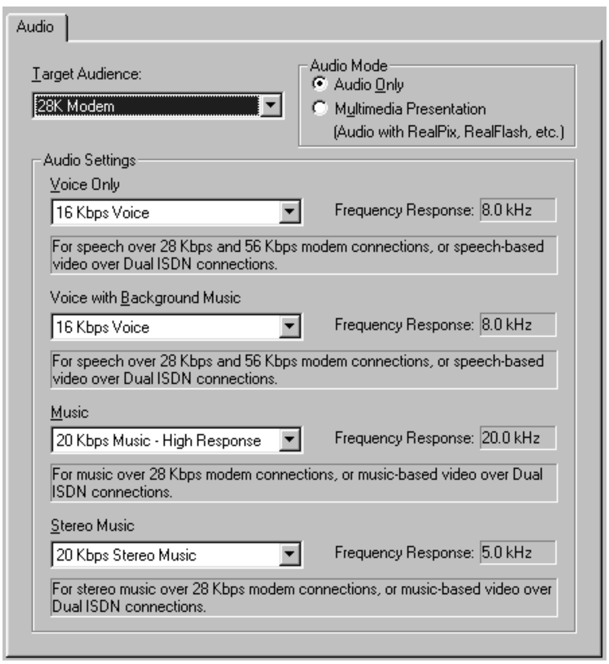
clip quality is all important, though, you may want to keep the 96 Kbps stream to wring out as much fidelity as you can get.

Of course, you don't have to eliminate the higher bit rate stream. In some cases, you may want to eliminate an intermediate stream. Take a look at the "voice and music" column of Table 2. To cut down the clip size, you might leave out the 56 Kbps modem and 64 Kbps single ISDN audiences. Users at these speeds would then get the 16 Kbps clip, the same one 28.8 Kbps modem users get. The clip would have a 16 Kbps and a 64 Kbps stream for higher speeds. As you see, reducing SureStream file size is a balancing act based on which audiences are important to you.

Encoding with the RealAudio Multimedia Defaults

Table 2 on page 7 shows you which RealAudio codecs RealProducer uses as its *standard default* settings. RealProducer has another set of defaults, though, that lower a RealAudio clip's streaming bandwidth so you can combine it with other clips in a multimedia presentation. To use the multimedia defaults, choose the menu command **Options>Target Audience Settings>for RealAudio Clips...**. This displays the dialog shown in Figure 2.

Figure 2: RealAudio Target Audience Dialog



Initially, the radio button **Audio Only** is selected. This sets RealProducer to use RealAudio's standard defaults given in Table 2. If you click **Multimedia Presentation**, RealProducer uses lower-bandwidth RealAudio codecs for each combination of audience and music type. This keeps RealAudio from overloading the connection when played in parallel with another clip in a SMIL presentation. These settings stay in effect until you click the **Audio Only** button to return RealProducer to the standard defaults.

Table 3 shows the results of encoding a RealAudio clip with the multimedia defaults. With these defaults turned on, encoding voice-only audio for a 28.8 Kbps modem produces an 8.5 Kbps clip, for example. The standard default given in Table 2 is a 16 Kbps clip. In most cases, a clip using the multimedia defaults consumes about half the standard default bandwidth. The "Top Speed" column in Table 3 shows the total usable bandwidth for each connection. Your multimedia presentation should not exceed this number.

Table 3: RealAudio Multimedia Default Bit Rates

<table>
<tr><th>Target Audience</th><th>Top Speed</th><th>Voice Only</th><th>Voice and Music</th><th>Mono Music</th><th>Stereo Music</th></tr>
<tr><td>28.8 Kbps modem</td><td>20 Kbps</td><td>8.5 Kbps</td><td>8.5 Kbps</td><td>8 Kbps</td><td>8 Kbps</td></tr>
<tr><td>56 Kbps modem</td><td>34 Kbps</td><td rowspan="2">16 Kbps</td><td>16 Kbps</td><td>20 Kbps</td><td>20 Kbps</td></tr>
<tr><td>64 Kbps single ISDN</td><td>45 Kbps</td><td rowspan="6">32 Kbps</td><td rowspan="3">32 Kbps</td><td rowspan="3">32 Kbps</td></tr>
<tr><td>112 Kbps dual ISDN</td><td>80 Kbps</td><td rowspan="5">32 Kbps</td></tr>
<tr><td>Corporate LAN</td><td>150 Kbps</td></tr>
<tr><td>256 Kbps DSL/cable</td><td>225 Kbps</td><td rowspan="3">64 Kbps</td><td>64 Kbps</td></tr>
<tr><td>384 Kbps DSL/cable</td><td>350 Kbps</td><td rowspan="2">96 Kbps</td></tr>
<tr><td>512 Kbps DSL/cable</td><td>450 Kbps</td></tr>
</table>

Tip

When you use RealProducer Basic, the same audience choice strategies discussed in "Choosing SureStream Rates with RealProducer Basic" on page 7 apply to using the multimedia defaults.

When to Use the Multimedia Defaults

Using the multimedia defaults is crucial for delivering multliclip presentations at slow connection speeds. Suppose you want to deliver a combined RealPix slideshow and RealAudio narration over a 28.8 Kbps modem. For this modem speed, presentations should not exceed 20 Kbps. If

you use the standard RealAudio settings, you get a 16 Kbps RealAudio clip, leaving just 4 Kbps for RealPix. That's inadequate for most slideshows. If you use the RealAudio multimedia settings, though, you get an 8.5 Kbps clip. That nearly triples the amount of bandwidth for RealPix to 11.5 Kbps.

At faster speeds, though, it's not always necessary to encode RealAudio at the multimedia rate for a multiclip presentation. Just make sure your RealAudio clip uses a lot of SureStream streams, and let each RealPlayer determine which stream to play. When clips play together, RealPlayer evaluates their separate requirements, assesses its own connection speed, and determines if it can juggle all the clips together. With a SureStream clip, RealPlayer automatically picks the highest bit rate stream it can handle while still keeping all the other balls in the air.

For example, suppose you have a RealPix clip at 12 Kbps. You could have it play in parallel with a RealAudio mono music clip encoded with the standard defaults for 28.8 Kbps and 56 Kbps modems. Table 2 on page 7 shows that you get two audio streams, one at 20 Kbps and one at 32 Kbps. The combined RealPix and RealAudio presentation is too fast for a 28 Kbps modem. For a 56 Kbps modem, though, RealPlayer chooses the 20 Kbps RealAudio stream, putting the presentation at 32 Kbps (20 Kbps for RealAudio plus 12 Kbps for RealPix). Faster connections get the 32 Kbps RealAudio stream.

In some cases the multimedia defaults are the same as the standard defaults, so it doesn't matter which set of defaults you use. For DSL/cable modem connections, for example, mono music is recorded at 64 Kbps regardless of which defaults you use. The same is not true for stereo music with the 256 Kbps DSL/cable modem audience, though. The standard default is 96 Kbps while the multimedia default is 64 Kbps.

Even when the rates differ between the standard and multimedia defaults, you may want to stick with the standard defaults when you have bandwidth to burn. Look at the voice-only speeds for the DSL/cable modem settings, for instance. The multimedia rates are 32 Kbps, while the standard rates are 64 Kbps. But the presentation top speed is a whopping 225 to 450 Kbps. When pairing RealAudio with a RealPix clip that streams at 40 Kbps, for example, you can encode your soundtrack with standard defaults, getting a 64 Kbps RealAudio rate. At a combined speed of 104 Kbps, your presentation will have plenty of bandwidth to spare on DSL and cable modem connections.

RealAudio, it's important to grasp, is the most *in*flexible media type in terms of bandwidth use. Bandwidth choices are in a stairstep model: 20 Kbps, 36 Kbps, 44 Kbps, and so on, with no in-between choices. So when you plan to

use RealAudio in a multimedia presentation, decide which bandwidth or bandwidths you want the audio clip to use. Then create your other clips to stream within the bandwidth that's left.

Tip

Remember not to count on having the connection's full speed (such as 28.8 Kbps) available. Always use as your maximum the top speed given in Table 3.

Changing RealAudio Defaults

If you own RealProducer Plus, you can choose exactly which RealAudio codec RealProducer uses for any combination of audience and audio type. You can't do this with RealProducer Basic, however. So if you want to become the ultimate power user, you'll need to upgrade.

To change the defaults, choose **Options>Target Audience Settings>for RealAudio Clips...** to display the dialog shown in Figure 2 on page 9. Click the **Audio Only** radio button to change the standard codecs, or the **Multimedia Presentation** button to change the codecs used for clips combined with other media. Next, select the connection speed in the **Target Audience** pull-down menu. The four pull-down menus in the remainder of the dialog let you set the target audience's codec for each type of audio:

- voice
- voice with music
- mono music
- stereo music

Because you can change any RealAudio default codec choice, there's nothing immutable about the target audience and audio type combinations. When you encode a clip, selecting voice-only audio for 28.8 Kbps modems might give you a stereo music clip for DSL connections because you've changed the defaults. This makes changing the defaults a powerful tool, but also a potentially confusing one if you forget how you've adjusted codec selection. Keep in mind, too, that you can choose between two distinct sets of defaults by clicking either the **Audio Only** or the **Multimedia Presentation** radio button.

Tip

Click **Restore Defaults** to return all settings to their defaults. This affects both the standard and the multimedia default settings.

Additional Information

The statistics that RealProducer shows after creating a clip help you verify that the clip is encoded the way you want. See "Viewing Statistics" on page 35.

Changing RealAudio Defaults for Different Bandwidths

The most obvious reason to change the RealAudio defaults is to modify the streaming speed for a certain target audience and audio type. This is most useful for multiclip presentations. As noted above, you might not want to use RealAudio's reduced multimedia rates at higher connection speeds because the standard defaults leave plenty of bandwidth for other clips. (That's rarely true at slow speeds, though.) So you might leave the multimedia defaults for slow connection speeds at their presets, and use faster codecs for higher connection speeds.

Even if you want reduced RealAudio bandwidths at higher speeds, changing the multimedia defaults lets you target exactly which codec to use. Look at the mono music rates for the LAN and DSL/cable modem connections listed in Table 2 on page 7 and Table 3 on page 10. The streaming rate with the standard defaults is a 64 Kbps. The multimedia rate weighs in at half that size: 32 Kbps. Between those two is a 44 Kbps mono music codec. If you're creating a presentation at 100 Kbps, and you want the best audio possible, you might bump the multimedia rate up to 44 Kbps as long as 56 Kbps is enough for the other clips.

Note

Always keep in mind the top streaming speed for connections as shown in Table 3 back on page 10.

Encoding High Response Music

The 20 Kbps and 32 Kbps mono music codecs both come in two flavors. RealProducer by default uses the "high response" versions, which are the better codecs for most situations. But you can also use the "normal response"

versions. Table 4 lists the high response codecs and their normal response twins.

Table 4: RealAudio High and Normal Response Codecs

Codec	Speed	Frequency Response
20 Kbps Music	20 Kbps	10 kHz
20 Kbps Music—High Response	20 Kbps	20 kHz
32 Kbps Music	32 Kbps	16 kHz
32 Kbps Music—High Response	32 Kbps	20 kHz

The high response codecs cover a larger frequency spectrum than the normal response versions. As you can see with the 20 Kbps codecs, the high response version has twice the range as the normal codec. This means it provides crisper sound and is better at capturing high frequencies. With symphonic music, for example, the high response codec gets more of the flute and piccolo. It can produce more distortion than the normal response codec with voices and loud sounds such as drums, though.

If you're encoding music with a diverse range of frequencies, stick with the high response codecs. If you notice distortion, compare your results with a clip that uses the normal response codecs. The best tool for determining which codec to use is your ear. Listen carefully for minute differences in how the clip sounds. It also helps to have other people listen. Our own ears have different frequency responses, too.

Making a Mono Clip from Stereo Input

Take a look back at the mono and stereo music settings in Table 2 on page 7. Until you get to the high speed connections, the mono and stereo bit rates are the same. What isn't the same, though, is the quality. Mono music has one channel that gets sent to both speakers. Stereo music has separate channels for the left and the right. That means a stereo version of an audio clip holds twice as much data as a mono version. Yet RealProducer makes stereo and mono clips the same size. How can it do this?

The answer lies in the codecs' frequency responses. Remember, RealAudio is a lossy compression scheme that throws out data. The stereo codec squeezes both channels down to the same size as the mono codec by throwing out more data for each channel. This makes its frequency response lower. To put it simply, a stereo clip doesn't represent all frequencies as accurately as a mono

clip. You'll hear the two channels, but they may not sound as crisp as a mono channel.

The lower frequency response is not necessarily a bad thing. If the music falls within the codec's response range, the mono and stereo clips won't differ greatly (except with stereo you get the two channels). If a stereo clip sounds muddy, try encoding the audio source as a mono clip. RealProducer is smart about the stereo-to-mono conversion. It doesn't simply throw out a channel as some audio programs do. Instead, it blends the two channels into the mono channel. This is called "panning to the middle." If it didn't do this, a musical instrument recorded on only the left or right channel might simply vanish from the recording!

Tip

If you're a pro or budding audiophile, run the stereo-to-mono conversion in your audio editing program so you have full control over the result.

Additional Information

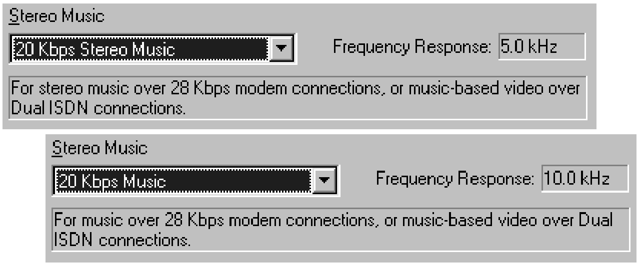
"RealAudio Codec Reference" on page 16 lists each RealAudio codec's frequency response.

Using Mono at Low Speeds, Stereo at High Speeds

Stereo encoding steps into the limelight at higher bandwidths. The higher the streaming speed, the better the stereo frequency response. You may find that the best compromise is to provide mono music at low bandwidths, stereo music at high bandwidths. You can do this by producing two different versions of your audio file. Or you can change the RealProducer Plus defaults to encode stereo music as mono at low speeds and as stereo at high speeds where the stereo codec frequency response is greater.

If you stick with the stereo codecs at higher connection speeds, SureStream clips are stereo when bandwidth is plentiful, mono when it's sparse. If the clip has to downshift from high to lower speeds, the music shifts from stereo to mono, but the sound stays brighter. You might change the 28.8 and 56 Kbps default targets for stereo music to use the 20 Kbps and 32 Kbps mono codecs, for example. Figure 3 illustrates changing the 28.8 Kbps stereo music target from stereo to mono. Note how the frequency response doubles.

Figure 3: Mono Encoding for Stereo Audio at for 28.8 Kbps Modems



Additional Information

"Changing RealAudio Defaults" on page 12 explains how to change the default codec selections. "RealAudio Codec Reference" on page 16 lists the frequency response of each codec.

RealAudio Codec Reference

This section provides a reference for all RealAudio codecs used by RealProducer, broken down into separate tables for voice, mono music, and stereo music codecs. There are no separate voice-with-music codecs. When you choose voice with music, RealProducer uses a voice codec. Each table lists each codec's optimum sampling rate and frequency response.

Using a codec's optimum sampling rate in your audio source file ensures that the audio stays synchronized with other media and prevents pitch shifting in audio resampling. Audio quality degrades if you use lower than the optimum sampling rate. If you use a higher sampling rate, it is best to use a multiple of the optimum rate. If the optimum rate is 8 kHz, for example, use a higher rate of 16 kHz or 32 kHz. When in doubt, use a CD-quality sampling rate of 44.1 kHz.

The frequency response column lists the codec's frequency response in Kilohertz. A codec with a higher frequency response reproduces a wider range of sound than a codec with a lower response. A measure of codec quality, the frequency response does not affect how you produce audio. RealAudio encoding always results in a clip of equal or lower quality than the original audio. If the original audio has an 8 kHz frequency response, encoding it with a codec that has a frequency response of 10 kHz produces a clip that still has a response of 8 kHz.

Additional Information

The audio preparation chapter of *RealSystem G2 Production Guide* has an expanded list that covers all RealAudio codecs, including obsolete codecs no longer used by RealProducer.

Voice Codecs

RealProducer uses a voice codec when you encode a voice-only or voice-with-music clip. The lowest-speed voice codec normally used with RealAudio is 16 Kbps. The lower-speed codecs are used as "duress" streams in SureStream clips. They're also used to encode soundtracks for low-bandwidth RealVideo clips.

Table 5: RealAudio Voice Codecs

RealAudio Codec	Sampling Rate	Frequency Response
5 Kbps Voice	8 kHz	4 kHz
6.5 Kbps Voice	8 kHz	4 kHz
8.5 Kbps Voice	8 kHz	4 kHz
16 Kbps Voice	16 kHz	8 kHz
32 Kbps Voice	22.05 kHz	11 kHz
64 Kbps Voice	44.1 kHz	20 kHz

Mono Music Codecs

As with the voice codecs, the lowest-speed mono music codec normally used with RealAudio is 16 Kbps. The lower-speed codecs are used as "duress" streams in SureStream clips, and to encode soundtracks for low-bandwidth RealVideo clips. When there are two versions of a codec, RealProducer uses the high response version by default.

Table 6: RealAudio Mono Music Codecs

RealAudio Codec	Sampling Rate	Frequency Response
6 Kbps Music	8 kHz	3 kHz
8 Kbps Music	8 kHz	4 kHz
11 Kbps Music	11.025 kHz	5.5 kHz
16 Kbps Music	22.05 kHz	8 kHz
20 Kbps Music	22.05 kHz	10 kHz

(Table Page 1 of 2)

Table 6: RealAudio Mono Music Codecs (continued)

RealAudio Codec	Sampling Rate	Frequency Response
20 Kbps Music—High Response	44.1 kHz	20 kHz
32 Kbps Music	44.1 kHz	16 kHz
32 Kbps Music—High Response	44.1 kHz	20 kHz
44 Kbps Music	44.1 kHz	20 kHz
64 Kbps Music	44.1 kHz	20 kHz

(Table Page 2 of 2)

Stereo Music Codecs

The slowest stereo codec is 20 Kbps. Stereo codecs don't go lower than that because they would not have enough frequency response for adequate sound. You can always encode stereo music with a lower-speed mono codec, though. RealProducer converts the music to mono for you. Don't encode mono music with a stereo codec, though. You don't get stereo music, just a lowered frequency response.

Table 7: RealAudio Stereo Music Codecs

RealAudio Codec	Sampling Rate	Frequency Response
20 Kbps Stereo Music	11.025 kHz	5 kHz
32 Kbps Stereo Music	22.05 kHz	8 kHz
44 Kbps Stereo Music	22.05 kHz	11 kHz
64 Kbps Stereo Music	44.1 kHz	16 kHz
96 Kbps Stereo Music	44.1 kHz	20 kHz

Working with the RealVideo Codec

A video consists of two parts: the visual track and the soundtrack. For the soundtrack, RealProducer uses the RealAudio codecs. For the visual track, it uses a RealVideo codec. It packages both tracks in a single RealVideo file that, like RealAudio, uses the file extension `.rm`. So keep in mind that everything discussed about RealAudio clips in "Working with RealAudio Codecs" starting on page 5 applies to the soundtrack of RealVideo clips.

Unlike RealAudio, RealVideo uses just one codec to compress a video's visual track for all bandwidths. For all practical purposes, the RealVideo codec is infinitely scalable. You can encode RealVideo at any speed you want, from 20

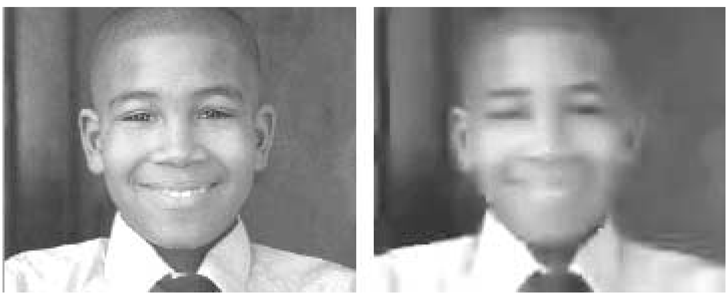
Kbps to hundreds of kilobits per second. Plus you can encode at precisely any bandwidth you choose, such as 65 Kbps, 89 Kbps, 117 Kbps, and so on.

Because RealVideo uses RealAudio for audio, a specific amount of the clip's bandwidth goes to the soundtrack. The visual track then gets squeezed into the remaining bandwidth. Like RealAudio, RealVideo compression is "lossy," meaning RealProducer throws out video data when necessary as it encodes a clip. RealVideo does this intelligently to keep the clip's playback quality as high as possible.

RealProducer squeezes down clip size by, *in part*, reducing the video's frame rate. Most videos have a frame rate from 15 to 30 frames per second (fps). RealProducer dynamically adjusts this frame rate downward, keeping the rate up where there's a lot of movement, reducing it in slow scenes. If you follow good production techniques, your clips will typically stream over slow to medium-speed connections at 5 to 15 fps, depending on the video quality and the target audience. At higher speeds, you'll get 15 to 30 fps.

Another way RealVideo reduces the clip's streaming size is to throw out pixel data. A standard video clip stores information about each screen pixel in the video frame. Instead of storing unique values for each pixel, RealVideo stores data for pixel groups. When bandwidth is tight, RealProducer shoehorns pixels with slightly different RGB values into the same group. These pixels end up looking identical rather than nearly identical. This results in a loss of detail, which can distort the video. Figure 4 compares a smooth video with a version that has lost detail through encoding for a low bandwidth.

Figure 4: Smooth and Distorted Video



The different means of reducing video bit rate explain why RealProducer asks you whether you prefer smooth motion (a higher frame rate), better image quality (more detail), or a split of the difference. Your choice tells it how to squeeze down the file data. As it creates the RealVideo clip, RealProducer also uses some complex technology to soften the blow of the reduced frame rate

and pixel data. This helps smooth jerkiness and sand off the rough edges of the pixel blocks.

Encoding Clips with the RealVideo Defaults

RealProducer determines how to encode a RealVideo clip based on your selection of:

- target audience speed (dial-up modem, DSL/cable modem, and so on),
- audio type (voice, voice with music, mono music, or stereo music), and
- video quality preference (smooth motion, clear image, a combination of the two, or a slide show).

Table 8 shows the RealAudio codecs used for RealVideo clips based on connection speed and audio type.

Table 8: RealVideo Bit Rates

<table>
<tr><th rowspan="2">Target Audience</th><th rowspan="2">Top Speed</th><th colspan="4">RealAudio Rate</th></tr>
<tr><th>Voice Only</th><th>Voice and Music</th><th>Mono Music</th><th>Stereo Music</th></tr>
<tr><td>28.8 Kbps modem</td><td>20 Kbps</td><td rowspan="2">5 Kbps</td><td rowspan="2">6.5 Kbps</td><td rowspan="2">8 Kbps</td><td rowspan="2">8 Kbps</td></tr>
<tr><td>56 Kbps modem</td><td>34 Kbps</td></tr>
<tr><td>64 Kbps single ISDN</td><td>45 Kbps</td><td rowspan="2">8.5 Kbps</td><td rowspan="2">8.5 Kbps</td><td>11 Kbps</td><td>11 Kbps</td></tr>
<tr><td>112 Kbps dual ISDN</td><td>80 Kbps</td><td>16 Kbps</td><td>20 Kbps</td></tr>
<tr><td>Corporate LAN</td><td>150 Kbps</td><td rowspan="4">32 Kbps</td><td rowspan="4">32 Kbps</td><td>32 Kbps</td><td>32 Kbps</td></tr>
<tr><td>256 Kbps DSL/cable</td><td>225 Kbps</td><td rowspan="2">44 Kbps</td><td>44 Kbps</td></tr>
<tr><td>384 Kbps DSL/cable</td><td>350 Kbps</td><td>64 Kbps</td></tr>
<tr><td>512 Kbps DSL/cable</td><td>450 Kbps</td><td>64 Kbps</td><td>96 Kbps</td></tr>
</table>

The first column in Table 8 lists RealProducer's standard target audiences. For each audience, the "Top Speed" column shows the maximum streaming bandwidth. For 28.8 Kbps modems, for example, RealProducer creates a RealVideo clip at 20 Kbps. As it does when creating a RealAudio clip, RealProducer never uses the full connection bandwidth, reserving some bandwidth for overhead.

The remaining four columns show which RealAudio codec RealProducer uses in the RealVideo clip, depending on the audio type. Compare these rates to those in Table 2 on page 7. Table 2 shows that when you encode mono audio for a 28.8 Kbps modem, you get a 20 Kbps RealAudio clip. But Table 8 shows

that the same audio gets 8 Kbps of bandwidth when used in a video clip. Even at fast speeds, a RealVideo soundtrack gets about half the bandwidth it would receive when encoded solo as RealAudio. So, for example, a music video won't sound as good as the music alone.

The amount of bandwidth used for the visual track is the total rate minus the RealAudio soundtrack rate. For a 28.8 Kbps modem, the visual track gets 15 Kbps (20 Kbps minus 5 Kbps) when the audio is voice-only, 13.5 Kbps (20 Kbps minus 6.5 Kbps) when it's voice with music, or 12 Kbps (20 Kbps minus 8 Kbps) when it's music. A video with an audio narration might therefore look slightly better than one with a music soundtrack because more bandwidth goes to its visual track. At higher connection speeds, though, the differences even out. Many things, though, affect a video clip's quality, and even a music video can look good at 20 Kbps.

Additional Information

For more on video quality, see the video preparation chapter in *RealSystem G2 Production Guide*.

Note, too, that the lowest-speed stereo codec is 20 Kbps. Table 8 shows that stereo codecs 20 Kbps or faster are used only for RealVideo clips encoded at dual ISDN or faster speeds. When you encode for lower connection speeds, RealProducer converts stereo sound to mono because bandwidth is lacking for both the visual track and a stereo soundtrack. "Making a Mono Clip from Stereo Input" on page 14 explains more about this conversion process.

Changing the RealAudio Codecs used with RealVideo

If you have RealProducer Plus, you can change the RealAudio codecs RealProducer uses for video clips. This lets you increase or decrease the soundtrack bandwidth to emphasize or de-emphasize it in relation to the visual track. To change the defaults, choose **Options>Target Audience Settings>for RealVideo Clips...**, and click the **Audio** tab. The dialog is similar to the one used to modify RealAudio defaults, as shown in Figure 2 on page 9.

To change the RealAudio codec used with a certain RealVideo stream, select the target audience speed in the **Target Audience** pull-down menu. The four pull-down menus in the remainder of the dialog correspond to the audio type choices: voice, voice with music, mono music, and stereo music. For an audio type, simply select the appropriate codec from the pull-down menu. Click **OK** when you're finished.

Note

Each tab of the RealVideo preferences dialog has a **Restore Defaults** button that returns everything to the initial defaults. This affects all panes in the dialog, not just the one you're viewing.

When to Modify the Audio Bandwidth in RealVideo

The reasons for swapping RealAudio codecs given in "Changing RealAudio Defaults" on page 12 applies to changing the audio defaults for RealVideo as well. When you encode a video, you might want to change its stereo soundtrack to mono to get better frequency response at low bandwidths. Or you might want to use the 20 Kbps and 32 Kbps normal response codecs.

The primary reason to change the defaults, though, is to increase or decrease the bandwidth used by the video's soundtrack. In most cases, you won't want to decrease the soundtrack's bandwidth. As Table 8on page 20 shows, the defaults are low to start. This ensures that the visual track gets enough bandwidth while keeping audio quality acceptable. For high-speed connections, though, you may want to increase the soundtrack bandwidth to get better sound quality.

An example of when to increase audio bandwidth is a music video streamed at high speeds. At the 256 Kbps DSL/cable modem speed, you may want to use a 64 Kbps RealAudio codec instead of 32 or 44 Kbps, for instance. This boosts sound quality greatly without cutting too hard into video clarity and frame rate. If in doubt, try different options and determine by sight and sound which clip best balances audio and visual quality.

Modifying RealVideo's Frame Rate

The default, *maximum* frame rate for RealVideo clips is 15 frames per second (fps) for dial-up modem and ISDN audiences, 30 fps for LAN/DSL/cable modems. During encoding, RealProducer adjusts the frame rate based on the clip size, target audience speed, and emphasis on smoothness or visual clarity. One scene may be 7 fps, for example, while another is 10. A maximum of 15 fps means the frame rate may vary anywhere between 15 fps and 0.25 fps.

It's a good strategy to leave the maximum frame rates set to their defaults and let RealProducer cut rates down as needed. At low bandwidths, though, you may want to emphasize clarity. Instead of just choosing the RealProducer option for visual clarity, you could also set a slower maximum frame rate, say 5

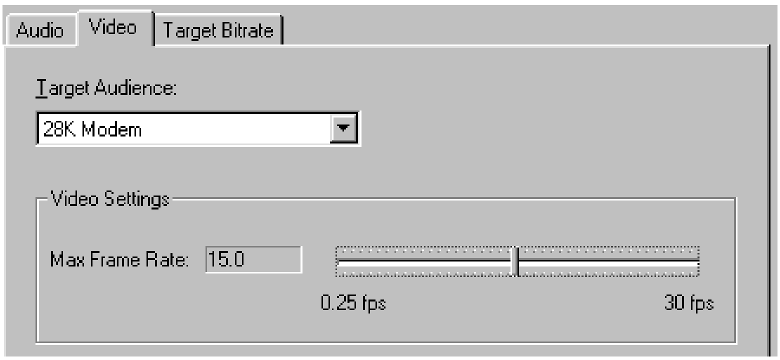
fps. This slows the frame rate even when it could go faster. The result is a sharper image, at the expense of some jerkiness due to the slower frame rate. Rarely would you want to increase the maximum frame rate unless you're also raising the streaming bit rate as described below.

Tip

To really slow things down and heighten visual clarity, you can just record the video as a slide show. You'll get about one frame a second without having to lower the frame rate setting.

You can change the maximum rate with **Options>Target Audience Settings>for RealVideo Clips...**. After giving this command, click the **Video** tab, which is shown in Figure 5. Select the connection speed in the **Target Audience** pull-down menu. Then click the slider button and move it between the minimum of 0.25 fps and the maximum of 30 fps. The chosen frame rate displays to the left of the slider.

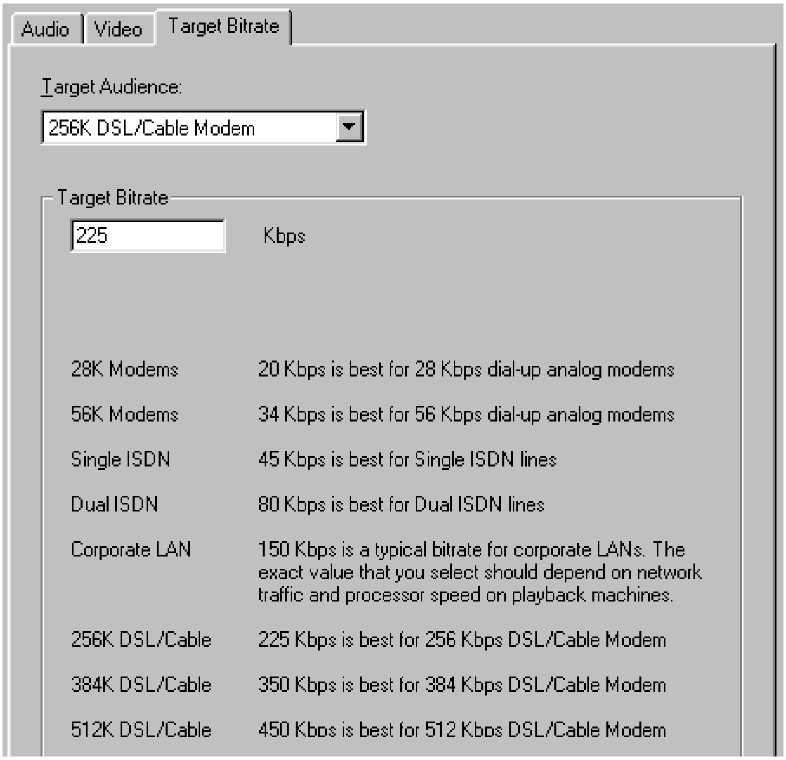
Figure 5: RealVideo Maximum Frame Rate Dialog



Changing RealVideo's Target Bit Rate

Unlike RealAudio codecs, which have fixed bit rates, the RealVideo codec can squeeze the visual track into any bandwidth. Only the soundtrack bit rate is fixed, based on the RealAudio codec used. To change RealVideo's total streaming speed, select **Options>Target Audience Settings>for RealVideo Clips...**. Then click the **Target Bitrate** tab, which is shown in Figure 6. This tab lets you change the total RealVideo bit rate by selecting the connection speed in the **Target Audience** pull-down menu, and entering a bit rate in Kilobits per second in the **Target Bitrate** field. Click **OK** when you've made your changes.

Figure 6: RealVideo Target Bit Rate Dialog



The main reason to lower the streaming bit rate of a RealVideo clip is to combine it with another clip at lower bandwidths. But remember that lowering the RealVideo bit rate affects its quality. Video is hungry for speed, and you need to leave enough bandwidth for RealVideo to function well. You generally shouldn't need to raise the bit rates because they are as high as they can safely go for each target audience. The following sections give tips for handling RealVideo bandwidth settings.

Additional Information

See the "Top Speed" column of Table 8 on page 20 for the recommended maximum bandwidth of a presentation streamed to various audiences.

Don't Lower RealVideo Speeds for 28.8 Kbps Modems

Don't touch the setting for 28.8 Kbps modems. The default size of 20 Kbps is the smallest package that can carry video. Lowering the size seriously degrades quality. So for slow modems, don't try to combine RealVideo with other clips,

except a lightweight RealText clip that consumes 1 Kbps or less. In this case, you don't need to lower the RealVideo bandwidth from 20 Kbps because RealText can easily sneak inside the extra 8.8 Kbps left for modem overhead.

Set Audio Bandwidth for a RealVideo Clip Separately

Whenever you lower RealVideo speed, the new bit rate includes the rate of the RealAudio codec set through the dialog's **Audio** tab. If you use a 32 Kbps RealAudio codec and set a total RealVideo bit rate of 36 Kbps, you won't get much for the visuals, maybe a frame every few seconds. You'll need to select a lower-bandwidth RealAudio codec, too.

There are no strict rules for the ratio of soundtrack bandwidth to total RealVideo bandwidth. The ratio can be anything depending on whether you want to emphasize audio or the visuals. As you can see from Table 8 on page 20, RealVideo normally allots no more than 1/4 of the total bandwidth to the soundtrack because the visual information contains much more data. At high bandwidths, the soundtrack can receive considerably less bandwidth in proportion to the visual track.

Additional Information

See "Changing the RealAudio Codecs used with RealVideo" on page 21.

Keep the LAN Manager Happy

When streaming video on an intranet, you can raise RealVideo's LAN connection bit rate because a local area network has total bandwidth much higher than 150 Kbps. However, LAN bandwidth is shared by everyone logged on. If lots of people view your RealVideo clips simultaneously, the LAN can bog down and no one is going to be happy, least of all the LAN manager. So in some cases, you may need to lower the LAN audience bit rate. What to set for the LAN rate depends on the LAN speed and average RealVideo use. Your LAN manager is the best person to determine this.

Encode for Different Target Audiences in Some Cases

Instead of lowering the bit rate for a target audience, you could just encode the clip using a slower target audience setting. This does not give you the precise control you have when choosing a new bit rate, but it may be adequate in many circumstances. Suppose you want to combine RealVideo with a 50 Kbps RealPix slide show for DSL/cable modem connections. You could lower RealVideo's lowest DSL rate by 50 Kbps, or simply encode the RealVideo clip

at the standard LAN speeds. The modified DSL setting gives you a 175 Kbps clip that is only marginally better than the 150 Kbps clip you get from the default LAN option.

Setting RealVideo Advanced Features

RealProducer has several options that affect how it encodes RealVideo clips. These options can improve clip quality, but also degrade it if used improperly. Some options also affect encoding speed. Choose **Options>Preferences** to display the preferences dialog, which has three panes of options.

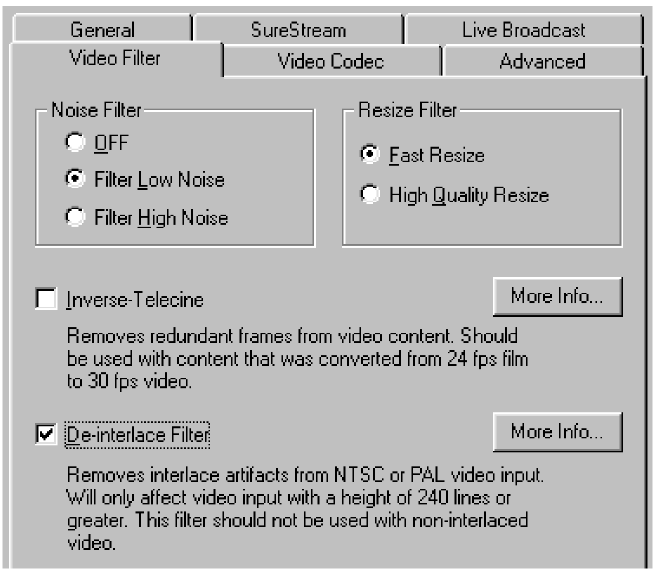
Note

Unless noted otherwise, all RealVideo options are compatible with each other and can be used when encoding video from a source file, a media device, or a live broadcast.

Using RealVideo Filters

The **Video Filter** pane of the preferences dialog has several filtering options you can use when encoding RealVideo. These filters remove artifacts that appear in the encoded clip because of the methods used to create the source video.

Figure 7: RealVideo Filter Preferences Dialog



Noise Filters

A by-product of poor quality in one or more links in the video production chain, video noise (which has nothing to do with the audio quality) can distort the encoded clip. These distortions are similar to the "snow" that often shows up in TV signals received over an antenna. The source of the noise is typically hardware, such as the video tape, capture card, or camera. Using professional-quality equipment and media helps eliminate video noise at the source. If your source video is high quality to start with, you won't need the noise filters.

Additional Information

See the video preparation chapter in *RealSystem G2 Production Guide* for tips on producing high-quality video.

Low Noise Filter

If your video input has a small amount of noise, turn on the low-noise filter. Because it has a small impact on processing power and won't degrade a video's appearance, the low noise filter is safe to leave on at all times. It's better practice, though, to use it only when necessary.

High Noise Filter

If noise greatly distorts the source video, try the high noise filter. Although optimized for Pentium MMX machines, the high noise filter can nonetheless add 30% or more to the encoding time. It can also remove slight details, making highly textured surfaces look more smooth. For these reasons, never use the high noise filter when it is not needed.

Resize Filters

RealProducer's **Options>Video Settings...** command lets you resize video as you encode it. On the **Video Filter** tab of the preferences, you can select whether to do this as a fast resize or a high-quality resize. These resize options affect the video only when you make it smaller. They tell RealProducer to throw out video data using a quick method (fast resize), or through a complex analysis (high-quality resize).

A fast resize has a small impact on encoding time, but the resulting image may have some distortion. A high-quality resize results in a superior image, but it may double or triple the encoding time because it carefully analyzes the video source before resizing. Because of its impact on speed, the high-quality resize filter is not recommended for broadcasts.

Inverse-Telecine Filter

The Inverse-Telecine filter is for cinematic film that was transferred to NTSC video, whether the VHS or BETA version. Film is usually photographed at 24 frames per second (fps), whereas the NTSC standard is 30 fps. The film-to-video conversion (called “telecine”) duplicates some frames to bring the film input up to the NTSC frame rate. American theatre-release films transferred to video, for example, undergo the telecine process.

Use the inverse-telecine filter when encoding NTSC video that was transferred from film and has a frame rate of 25 to 30 fps. (The filter is not necessary if the video frame rate is below 25 fps.) The filter strips out redundant frames, letting RealProducer focus on image quality. This improves the clip’s overall look. Although the inverse-telecine filter is safe to use on all input, it slows performance marginally and should be used only when the source is NTSC video that originated from film.

Note

PAL video, which is widely used in Europe, does not require the inverse-telecine filter because the conversion from 24 fps film to 25 fps PAL does not use the telecine process.

De-interlace Filter

The de-interlace filter removes jaggedness in interlaced NTSC or PAL video. A video camera running at 30 frames per second captures the odd-numbered lines of a frame in 1/60th of a second, and the even-numbered lines in the next 1/60th of a second. It then interlaces the two to create the frame. Because half the frame’s lines are captured a fraction of a second later than the other half, fast-moving objects may appear jagged, the result of the object advancing slightly within 1/60th of a second. Figure 8 illustrates this jaggedness in a detail of an interlaced video.

Figure 8: Jaggedness in an Interlaced Video (detail)

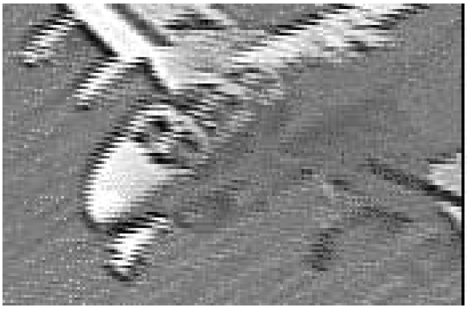
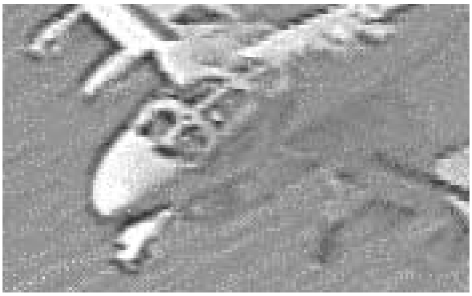


Figure 9 shows the jaggedness removed with the de-interlace filter.

Figure 9: Jaggedness Removed with the De-interlace Filter (detail)

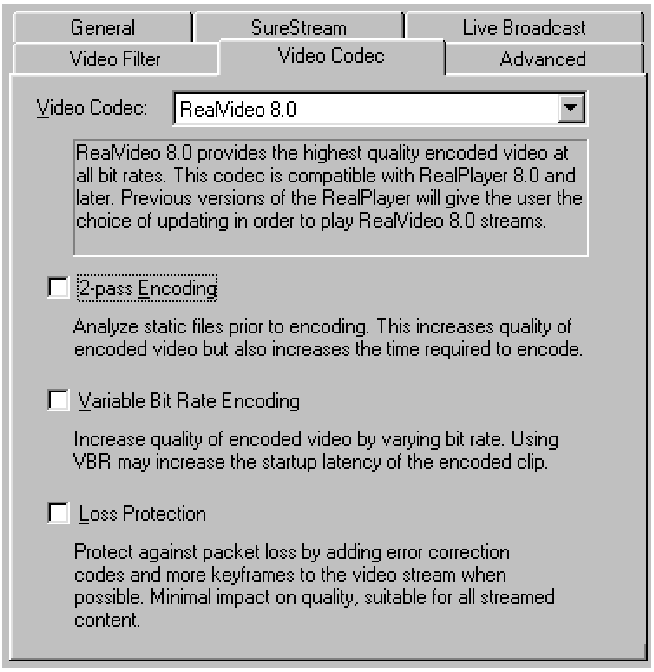


Optimized for Pentium MMX processors, the de-interlace filter has a modest impact on encoding speed, but is useful only for interlaced source video that is 240 lines or higher. Typical source video used for television is 480 lines high. If you digitize the video with a video capture card that captures at 240 lines high or less, the card throws out either the odd or the even lines, de-interlacing the video itself. The de-interlace filter is safe to leave on, though, because RealProducer never applies it to a video less than 240 lines high.

Setting RealVideo Codec Features

The **Video Codec** pane of the preferences dialog includes several encoding options you can use with RealVideo. It also lets you select which RealVideo codec encodes your clips. These options affect the quality of RealVideo clips by modifying RealProducer's encoding methods.

Figure 10: RealVideo Codec Preferences Dialog



RealVideo Codec

The top of the **Video Codec** pane has a pull-down menu that lets you choose between three RealVideo codecs. The default choice is the RealVideo 8.0 codec, but you can also choose one of two RealVideo G2 codecs. The codec you select encodes all of a clip's SureStream streams. You cannot encode half the streams with the RealVideo 8.0 codec, for example, and the other half with a RealVideo G2 codec.

RealVideo 8.0

The default RealVideo 8.0 codec results in visual quality markedly superior to that of the RealVideo G2 codecs. It requires more processing power, though, so encoding a clip with it takes longer than encoding the clip with a RealVideo G2 codec. RealPlayer 8.0 and higher can play RealVideo 8.0 clips. Earlier RealPlayers attempting to view a RealVideo 8.0 clip are prompted to auto-upgrade to RealPlayer 8.0. RealNetworks recommends using this codec unless you need faster encoding performance during broadcasts, or must reach older versions of RealPlayer.

Note

Most users upgrade their RealPlayers soon after a new version is released, so it is generally safe to use the RealVideo 8.0 codec.

RealVideo G2 Codecs

The RealVideo G2 codecs are older codecs used by RealProducer G2 and RealProducer 7. These codecs are faster than RealVideo 8.0, but their visual quality is poorer. Use them for faster encoding during broadcasts, or if you must reach RealPlayers that cannot upgrade to Release 8:

- RealVideo G2 with SVT

  The RealVideo G2 with SVT codec is compatible with RealPlayer version 6.0.6 or higher. RealPlayers with lower version numbers are prompted to auto-upgrade to the latest RealPlayer before viewing the clip.

- RealVideo G2

  The RealVideo G2 codec without SVT is compatible with RealPlayer G2, 7, and 8. It is not compatible with RealPlayers version 5.0 or earlier.

How SVT Works

The RealVideo 8.0 and RealVideo G2 with SVT codecs include Scalable Video Technology (SVT). This technology removes a stumbling block of streaming media by letting you encode clips at high frame rates for fast machines without concern that slower machines might not keep up. SVT even works when delivering clips with Web servers.

RealVideo's variable frame rate means one scene may be encoded at 7 fps, while another is at 15 fps. High frame rates take a lot of processing power to decompress. Although fast PCs handle high frame rates well, slower PCs may have trouble. With SVT, RealPlayer can lower the frame rate "on the fly" to keep the PC's CPU from sputtering. So although a certain scene is encoded at 15 fps, it may play on some RealPlayers at 8 fps, for example, if those RealPlayer machines lack the power to decompress 15 fps.

Loss Protection

The loss protection feature adds error correction data to RealVideo streams, helping them maintain quality in lossy environments. RealProducer adds as much error correction data as it can without taking away any video quality. Although you'll get more benefit from this feature when streaming on the Internet than on an intranet, it is safe to leave loss protection on for all

encoded content. This feature has a negligible impact on RealProducer's encoding speed.

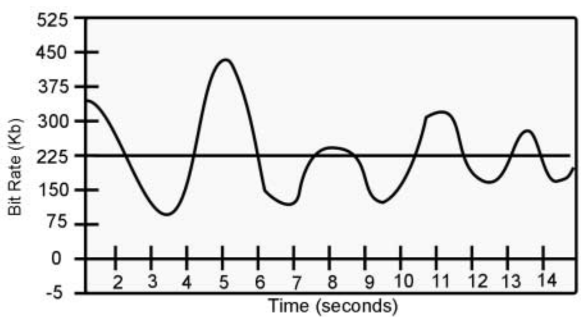
Variable Bit Rate Encoding

Variable bit rate (VBR) encoding varies a RealVideo clip's *playback* bit rate, giving more bandwidth to scenes that are hard to compress, and less to scenes that are easy. Compatible with SureStream and broadcasting, VBR encoding generally provides superior video quality to constant bit rate (CBR) encoding, which RealProducer uses if you do not select the VBR option. VBR makes the most difference in videos that have a mix of high-action and low-action scenes because it can steal bandwidth from low-action areas to give to high-action areas. This is particularly useful for improving video quality at low bit rates.

To illustrate how VBR encoding works, suppose you encode a video for a DSL/cable modem audience at 225 Kbps. With CBR, the video gets 225 Kilobits of encoded data each second. With VBR, though, each second of video may be encoded at a different rate. One second may have 150 Kilobits of data, for example, while another second has 300 Kilobits. The VBR clip will have a *streaming* bit rate of 225 Kbps, though, just like a CBR clip. So you do not need to worry that a VBR clip will underuse or overload a connection's bandwidth.

Think of the playback bit rate of a VBR clip as a waveform with peaks and troughs, as illustrated in Figure 11. RealProducer preloads the extra Kilobits required to play upcoming peaks in the preceding troughs. So one low-action scene that requires 150 Kilobits of data per second actually streams at 225 Kbps, carrying with it an extra 75 Kbps that RealPlayer uses in an upcoming fast-action scene. The flat line at 225 Kilobits in Figure 11 represents this streaming bit rate.

Figure 11: Variable Bit Rate Peaks and Troughs



A VBR-encoded video that starts with a high action scene needs a spike of bandwidth right away. If there are no preceding troughs to carry this data, RealPlayer has to buffer the clip longer. That means it may take the VBR clip longer than a CBR clip to start playing back. RealProducer lets you set the maximum time RealPlayer needs to buffer the clip, though, to ensure that the initial buffering time remains acceptable.

Additional Information

See "Setting VBR Maximum Start-up Latency" on page 34 for information on changing RealPlayer's initial buffering time.

Two-Pass Encoding

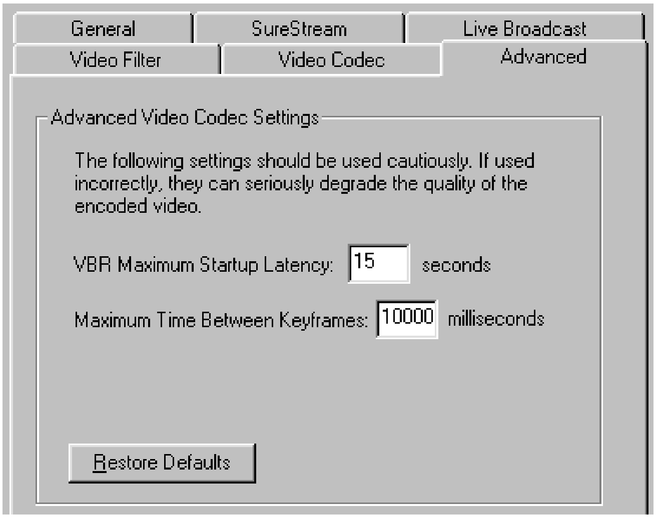
With two-pass encoding, which is used only when encoding from a digitized source file, RealProducer runs through the entire source video once to gather information about how best to encode the streaming clip. It then makes a second pass to encode the streams. Two-pass encoding can substantially increase clip quality, but it requires more encoding time. The first pass takes about as long as it would to encode the source file for one target audience.

Although two-pass encoding helps when you use constant bit rate encoding, it provides greater benefit for variable bit rate (VBR) encoding, which is described above. With two pass encoding, RealProducer can analyze the entire video file to determine how best to vary the playback bit rate through the length of the clip. Without two-pass encoding, RealProducer sequentially analyzes small sections of the source file during encoding, creating a string of VBR sections within the clip.

Using RealVideo Advanced Codec Features

The **Advanced** pane of the preferences dialog has options that affect RealVideo encoding. You should change these only with caution because they can greatly affect how well the clip streams.

Figure 12: RealVideo Advanced Codec Preferences Dialog



Setting VBR Maximum Start-up Latency

The **VBR Maximum Startup Latency** field affects only RealVideo clips encoded with a variable bit rate (VBR), which is described in "Variable Bit Rate Encoding" on page 32. A VBR video that starts out with a high-speed scene needs more initial buffering because the first scene is encoded at a playback bit rate higher than the audience connection speed. Because RealServer can't stream the scene *faster* without overloading the connection bandwidth, it streams it *longer* to deliver the extra Kilobits needed.

The latency field determines how long RealPlayer viewers may have to wait before a VBR video starts back. The default value of 15 seconds means that no matter how complicated the video's first scene, RealProducer will encode it so that it requires no more than 15 seconds to start playing. The field sets a maximum value only, and RealVideo VBR clips may start playback sooner. You can change the maximum to a whole value from 5 seconds to 25 seconds.

The value of 15 seconds is RealNetworks' recommendation. Keep in mind that this represents 15 seconds of clip data buffering, and does not include the time it takes to launch RealPlayer, find the host RealServer, send the request, and receive the host's response. If a low start-up time is critical, lower the latency time to 10 seconds, for example. For comparison, constant bit rate clips are encoded to have a latency of about 5 seconds. If initial image quality is crucial, you can raise the latency time, but this may cause restless viewers to stop the presentation before it begins playback.

Specifying Maximum Time Between Keyframes

RealProducer encodes full data for a frame in a keyframe. Successive frames encode just the data that describes how they vary from the preceding frame, starting with the keyframe. The **Maximum Time Between Keyframes** field is set initially to 10,000 milliseconds, meaning a constant or variable bit rate RealVideo clip has a keyframe at least every 10 seconds. The main reason to change the maximum rate is to lower it, although you should do this with caution. Lowering the rate generates more key frames for clips and provides several benefits:

- Minimizes distortion when streaming in a lossy environment.
- Improves RealPlayer's ability to seek to specific points in the RealVideo timeline.
- Adds flexibility for editing RealVideo clips through **File>Edit RealMedia File**. You have to cut a RealVideo clip at a keyframe, for example. More keyframes means more precise control over where the cut occurs.

Because keyframes encode much more data than other frames, though, lowering the maximum time between keyframes can lower the clip's image quality if you do not also raise the clip's streaming bandwidth. If you change the keyframe rate, test the clip quality to determine if the modification produced the desired results.

Additional Information

See "Changing RealVideo's Target Bit Rate" on page 23 for information on raising a clip's streaming bandwidth.

Viewing Statistics

After you record a clip, the confirmation dialog has a **View Statistics** button you can click to pull up a wealth of information about the clip. Even while RealProducer encodes the clip, you can watch statistics in real-time with **View>Statistics**. Although it's easy to pass these dry numbers by, they tell you a lot about your clip, including which speeds they're encoded for. The statistics dialog has up to three panes:

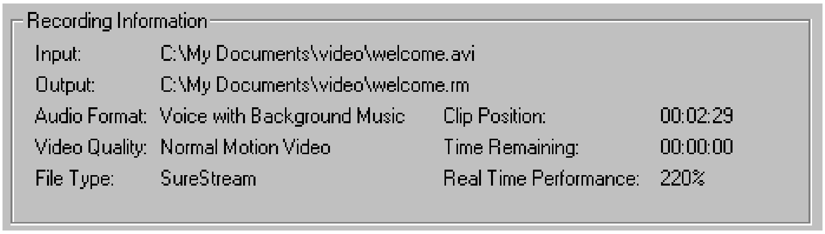
- The **General** pane appears only for RealVideo clips. It has combined statistics for the visual track and soundtrack.
- The **Audio** pane lists statistics for the encoded audio. It appears for RealVideo and RealAudio clips.

- The **Video** pane, which appears only for RealVideo clips, lists additional information about the visual track encoding.

Basic Recording Statistics

Figure 13 shows the basic recording information that appears at the top of the **General**, **Audio**, and **Video** panes of the statistics dialog. This example is for a SureStream RealVideo clip encoded with RealProducer Plus and targeting seven audiences.

Figure 13: Recording Information



As shown in Figure 13, the information at the top of each statistics pane indicates the basic choices made for audio format and video quality. When you view this information while encoding a file, the **Clip Position** field shows precisely where in the file's timeline RealProducer is encoding the clip. If it's two minutes and thirty seconds into the file, for example, **Clip Position** reads `00:02:30`. When you broadcast, this field is labeled **Duration**, and shows how long the stream has been broadcasted.

The **Time Remaining** field indicates how much longer RealProducer will take to finish the encoding. RealProducer may finish faster or slower depending if other computer processes use the CPU during the encoding. The **Time Remaining** field doesn't work when broadcasting or encoding a clip from a media device. In these cases, the encoding lasts until you turn the input off.

The **Real-Time Performance** field shows a percentage value that indicates overall codec performance. The lower the percentage, the faster the codecs are working. If RealProducer takes just under five minutes to encode a clip that lasts five minutes, the performance is around 100%. On a fast machine, performance will more likely be a lower value. If it takes four minutes to encode the five-minute clip, performance is around 80%. On a slow machine, the performance value may go above 100%. Taking six minutes to encode a five-minute clip, for instance, delivers a performance of around 120%.

Note

Performance is a measure of codec speed only. It does not take into account the time required for other activities such as disk reads and writes.

A performance over 100% is not a problem when you encode from a file. This just means it takes longer to encode the clip than to play the original file start to finish. Performance over 100% is a problem when you encode from an input device or broadcast a stream, however. It means RealProducer cannot encode the input as quickly as it comes in. For broadcasts, RealNetworks recommends a performance of 80% or less to ensure smooth operation. If necessary, you can boost performance by one or more means:

- Run RealProducer on a faster machine.
- Reduce the number of SureStream streams.
- Use the RealVideo G2 with SVT codec instead of the RealVideo 8.0 codec. (See "RealVideo Codec" on page 30.)
- Turn off filters that may slow encoding time:
  - High Noise Filter (see page 27)
  - Resize Filters (see page 27)
  - Two-Pass Encoding (see page 33)

Tip

Before broadcasting, carry out a test run to check how well RealProducer performs. In your test, RealProducer should send clip data to RealServer. RealServer does not need to broadcast the streams to more than a few RealPlayers, though.

General Statistics

The **General** statistics pane, the main part of which is shown in Figure 14, appears only for RealVideo clips. It lists the total bit rate, video bit rate, audio bit rate, frame rate, and buffer time for each stream in the clip. Figure 14 is for a SureStream RealVideo clip targeting seven audiences.

Figure 14: General Statistics Screen

RealPlayer G2 Streams

Target Audience	Total Bitrate	Video Bitrate	Audio Bitrate	Frame Rate	Buffer Time
512K DSL/Cable Modem	454.4 Kbps	422.4 Kbps	32.0 Kbps	15.0 fps	3.5 sec
384K DSL/Cable Modem	354.0 Kbps	322.0 Kbps	32.0 Kbps	15.0 fps	4.1 sec
256K DSL/Cable Modem	228.4 Kbps	196.4 Kbps	32.0 Kbps	15.0 fps	4.6 sec
Corporate LAN	152.5 Kbps	120.5 Kbps	32.0 Kbps	13.8 fps	4.9 sec
Corporate LAN	104.9 Kbps	72.9 Kbps	32.0 Kbps	9.9 fps	5.2 sec
Dual ISDN	81.4 Kbps	72.9 Kbps	8.5 Kbps	9.9 fps	5.3 sec
Dual ISDN	61.1 Kbps	52.6 Kbps	8.5 Kbps	5.6 fps	4.9 sec
56 Kbps Modem	34.7 Kbps	28.2 Kbps	6.5 Kbps	6.1 fps	5.1 sec
28K Modem	20.3 Kbps	13.8 Kbps	6.5 Kbps	5.7 fps	4.5 sec
28K Modem	15.3 Kbps	8.8 Kbps	6.5 Kbps	3.4 fps	4.7 sec
28K Modem	12.3 Kbps	5.8 Kbps	6.5 Kbps	0.1 fps	7.9 sec

Note the number of audience targets. Even though the clip is encoded for seven targets, 11 show up in this statistics dialog, which illuminates the inner workings of SureStream. Look at the two entries for Corporate LAN. They both use a 32 Kbps RealAudio codec, but the first has a total bit rate of 152.5 while the second has a 104.9 Kbps rate. The second stream is the duress stream that RealServer uses if the connection bogs down and it can't stream the full 152.5 Kbps of data. The audio stays the same, but as you can see from the **Frame Rate** column, the frame rate drops from 13.9 fps to 9.9 fps when RealServer downshifts.

Additional Information

For more on duress streams, read "Duress Streams" on page 4.

If you look at these statistics from top to bottom, you can see exactly how the clip compensates if it starts to lose bandwidth. Each time it needs to downshift, the clip reduces bandwidth consumption with one or a combination of three things:

1. Switching to a lower bandwidth RealAudio codec.
2. Switching to a visual track that has a lower frame rate.
3. Switching to a visual track that has reduced clarity.

The last column of the statistics pane lists the time in seconds that RealPlayer takes to buffer a stream before playing it. A value of 3.5, for example, means

RealPlayer starts to play the clip 3.5 seconds after it begins to receive clip data, assuming that the connection's bandwidth holds up and there's no network congestion. Ideally, you want buffering to be under 15 seconds. This buffering occurs only when the clip begins to play, not during downshifting or upshifting to other SureStream streams.

Audio Statistics

The **Audio** pane, shown in part in Figure 15, appears in the statistics dialog for RealVideo and RealAudio clips. It gives the frequency responses of the RealAudio codecs used to encode the clip. "RealAudio Codec Reference" on page 16 lists the frequency responses for all RealAudio codecs.

Figure 15: Audio Statistics Screen

RealPlayer G2 Streams

Audio Codec	Frequency Response	Real Time Performance
32 Kbps Voice	11.0 kHz	4%
8.5 Kbps Voice	4.0 kHz	14%
6.5 Kbps Voice	4.0 kHz	14%

Like the general real-time performance value shown at the top of each pane, the real-time performance value for each codec shows how quickly RealProducer created each RealAudio stream. Any value over 100% means RealProducer could not encode the track as fast as the data came in.

Note

As noted in "Basic Recording Statistics" on page 36, an overall performance of 80% or higher can cause problems when you encode live input.

Video Statistics

Illustrated partially in Figure 16, the **Video** pane displays only for RealVideo clips. It gives the video bit rate and frame rate for all streams, just like the **General** pane. In addition, it provides a quality index and information about real-time performance.

Figure 16: Video Statistics Screen

RealPlayer G2 Streams

Video Codec	Video Bitrate	Frame Rate	Quality Index	Real Time Performance
RealVideo 8.0	421.7 Kbps	15.0 fps	100	64%
RealVideo 8.0	321.2 Kbps	15.0 fps	100	64%
RealVideo 8.0	196.4 Kbps	15.0 fps	100	64%
RealVideo 8.0	120.1 Kbps	14.5 fps	100	61%
RealVideo 8.0	73.3 Kbps	12.2 fps	100	50%
RealVideo 8.0	52.7 Kbps	7.2 fps	100	29%
RealVideo 8.0	28.3 Kbps	3.5 fps	100	13%
RealVideo 8.0	13.9 Kbps	5.2 fps	100	20%
RealVideo 8.0	8.7 Kbps	3.0 fps	100	12%
RealVideo 8.0	5.5 Kbps	0.3 fps	100	1%

The quality index is useful only during live broadcasts. A value of 100 indicates the codec could keep up with the data coming in. A value below 100 means the computer does not have enough processing power to encode the stream in real-time. In this case, the codec compensates by dropping frames from the stream. The real-time performance value for each video encoding tells how quickly RealProducer created each visual track. Any value over 100% means RealProducer could not encode the track as fast as the data came in.

Note

As noted in "Basic Recording Statistics" on page 36, an overall performance of 80% or higher can cause problems when you encode live input.